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(71) Applicant (*for all designated States except US*): TELEFONAKTIEBOLAGET LM ERICSSON (publ) [SE/SE]; S-126 25 Stockholm (SE).

(72) Inventors; and
(75) Inventors/Applicants (*for US only*): MECKLIN, Tomas [FI/FI]; Laaksolahdentie 11 E, FIN-02720 Espoo (FI). HOLMQVIST, Hans [FI/FI]; Jokikaari 2, FIN-10300 Karjaa (FI). AULA, Juhani [FI/FI]; Karhunkierros 9, FIN-02400 Kirkkonummi (FI).

(74) Agent: BORENIUS & CO OY AB; Kansakoulukuja 3, FIN-00100 Helsinki (FI).

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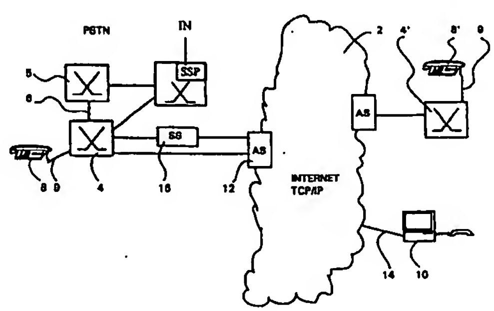

(57) Abstract

The invention relates to a method of setting up a connection between a first point within a circuit switched network and a second point within a packet switched network, wherein data is routed in the circuit switched network using signalling information and data is routed in the packet switched network using data network addresses. A translation table containing signalling information for a set of unique connections within the circuit switched network and respective data network addresses in the packet switched network is provided. For a connection initiated from said first point to said second point using signalling information, the signalling information is converted to a data network address using said translation table, and for a connection initiated from the second point to the first point using a data network address, the data network address is converted into signalling information using the translation table. The invention relates also to a telecommunication system and a signalling server for the same purpose.

CALL ROUTING IN COMMUNICATION NETWORKS

FIELD OF THE INVENTION

The present invention relates to routing of a call between networks, and more precisely, but not exclusively, to routing of a call between two or more telecommunication networks using different kind of addressing systems.

BACKGROUND OF THE INVENTION

At present most of telephone calls between two subscribers are accomplished by means of conventional circuit switched telephone systems. The oldest and hitherto largest telecommunications network in existence is the public switched telephone network (PSTN), which was for a long time the only bearer network available for telephony. During the recent years, however, the number of mobile telephone users using some of the various mobile telephone networks (public land mobile networks; PLMNs) providing mobility for the users thereof via radio connections has increased rapidly. Other bearer networks for voice transmission include arrangements such as integrated services digital network (ISDN), asynchronous transfer mode (ATM), frame relay and the Internet.

In order to be able to establish a call between a calling party (A-number) and a called party (B-number) in the circuit switched telephone systems, such as the PSTN, some predefined signalling and switching operations are required so that the call will become routed correctly between the A-number terminal and the B-number terminal. In the present telephone systems the call between the calling party A-number and the called party B-number is routed on basis of the B-number the calling party gives to the network system. In this connection signalling system No. 7 (SS7) by ITU-T can be named as an example of a modern signalling concept used in digital

circuit switched connections for common channel signalling (CCS). The usual parameters of the routing information will be described in more detail later in this specification.

In addition to voice traffic, i.e. the conventional telephony, the present communication networks are also capable of transmitting data. In most cases this is accomplished by means of packet switched data networks which convey the data in form of data packets. Data networks based on X.25 protocol (e.g. PSPDN; packet switched public data network) or the Internet are mentioned herein as examples of the packet switched data networks. From these the Internet can be defined as a global network using packet switched connectionless transfer mode and which is formed by the interconnection of thousands of subnetworks (e.g. various local or area networks; LANs, MANs etc.) that make the use of TCP/IP (transmission control protocol/Internet protocol) protocol suite and a common address structure. Unlike transfer techniques such as X.25, frame relay and the majority of ATM applications, the Internet is an end-to-end application extending all the way to the user terminal.

When establishing a connection between two terminals connected to the TCP/IP data network the routing is accomplished by means of an IP (Internet Protocol) address provided by the TCP/IP and indicating the desired destination terminal connected to the network. The source and the destination of the transmitted datagram are identified through use of fixed length IP addresses which are included within an IP header of the datagram. The length of an IP address depends on the used version, and is, for example, four octets in length in IPv4 (IP version 4) and 16 octets in IPv6 (IP version 6). The IP address consist of two parts: the network portion and the host computer portion.

In addition to data, the packet switched networks can also be used for transmitting sound, such as speech, between two communicating parties. When transmitting sound in data

network, the sound is digitised and transmitted in the digitised form as data packets. The digitised sound is then converted back into analogue form and represented to the listener at the receiving end.

The existing systems are also capable of transferring sound between terminals connected to different networks, e.g. to a local area network (LAN) and the Internet. In the prior art LAN-phone arrangements it is necessary to have a PBX (Private Branch Exchange) interfacing the LAN to the public telecommunication network used for the communication outside the LAN. The PBX has a specific routing table for the IP addresses of the terminals (i.e. LAN-phones) which are connected to the LAN. When making a call to a LAN-phone, the caller calls to the PBX, and thereafter the call is routed to the desired terminal by means of the routing table of the PBX. This kind of arrangement of the LAN networks cannot, however, be implemented in calls where the destination terminal or the originating terminal is connected directly to a conventional telecommunication networks, such as the PSTN.

SUMMARY OF THE INVENTION

The recent development of the communication systems is leading towards solutions where telephone networks and data networks are merged such that calls can be made from a telephone network to a data network and vice versa. However, a problem still not properly solved relates to the manner of routing calls from a network using one type of addressing system to a network using a different type of addressing system. This is the case e.g. in the PSTN to Internet routing, i.e. when the calling party is connected to a normal public circuit switched network utilising B-number for addressing purposes and the called party is connected to a data network using a different kind of addressing system, e.g. an IP address, or when a call between two PSTN subscribers is transmitted over an IP network.

It is an object of the present invention to overcome the disadvantages of the prior art solutions and to provide a new type of solution for routing calls between different networks.

Another object of the present invention is to provide a solution by means of which a calling party is allowed to make a call to a destination terminal beyond a network using different kind of addressing system in a manner similar to a call made to a destination terminal having an addressing system similar to that of the calling party terminal.

The objects are obtained by a method of setting up a connection between a first point within a circuit switched network and a second point within a packet switched network, wherein data is routed in the circuit switched network using signalling information and data is routed in the packet switched network using data network addresses, comprising steps of providing a translation table containing signalling information for a set of unique connections within the circuit switched network and respective data network addresses in the packet switched network, for a connection initiated from said first point to said second point using signalling information, converting the signalling information to a data network address using said translation table, and for a connection initiated from the second point to the first point using a data network address, converting the data network address into signalling information using the translation table.

According to further embodiments the signalling information parameters include at least one of the parameters of signalling identification field (SIF) in a message signalling unit (MSU), preferably at least one of originating point code (OPC), destination point code (DPC), circuit identification code (CIC), network indicator (NI), requesting country code (RCC). The signalling information may consist of a combination of originating point code (OPC), destination

point code (DPC), circuit identification code (CIC) and network indicator (NI). The packet switched data network is preferably the Internet, the addresses thereof being formed in accordance with IP address protocol. The signalling parameters and the data network address information can be stored in a signalling server and accomplishing the comparing of routing parameters and the address information in the signalling server. The call establishment signalling can be routed to the signalling server and the speech path is routed directly to the destination exchange.

According to a second aspect, the invention provides a telecommunication system comprising a circuit switched network wherein the routing of data is based on routing information parameters, a packet switched network wherein the routing of data is based on data network addresses, an access node between the networks, a signalling node including a record of the routing information parameters forming unique combinations for a set of connections and respective data network addresses, wherein the signalling node is arranged to associate the unique combinations of the routing information parameters with the data network addresses for obtaining a desired destination point for a call made from the circuit switched network to the packet switched network or vice versa.

According to a further aspect, the invention provides a signalling server including a translation table containing signalling information for a set of unique connections within a circuit switched telecommunication network, and respective data network addresses in a packet switched network, wherein the signalling server is arranged to, for a connection initiated from a first point to a second point using signalling information, convert the signalling information to a data network address using said translation table, and, for a connection initiated from the second point to the first point using a data network address, convert the data network address into signalling information.

The present invention provides a reliable and controllable manner for routing calls between terminals connected to different types of networks. The user of a public telecommunication network can transparently use normal telephone numbers instead of any other type of addresses than the ones the user is used to, e.g. normal telephone numbers when calling to a terminal which in reality is beyond a packet switched network. In addition, the invention enables mobile roaming over an IP network.

In the following the present invention and the other objects and advantages thereof will be described in an exemplifying manner with reference to the annexed drawings, in which similar reference characters throughout the various figures refer to similar features.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a schematic presentation of a communication system including a circuit switched telephone network and a packet switched data network and necessary gateway apparatus;

Figure 2 discloses main functions of SS7 layers;

Figures 3a and 3b are schematic illustrations of message parameters for a TUP message and an ISUP message, respectively;

Figure 4 is a schematic presentation of a table included in a routing server;

Figure 5 discloses a flow chart for the operation of one embodiment of the present invention; and

Figure 6 illustrates ISUP signalling sequences.

DETAILED DESCRIPTION OF THE DRAWINGS

Figure 1 is a schematical presentation of one communication network system comprising two different telecommunication networks, namely a PSTN 1 and the Internet 2. From these the PSTN is a public circuit switched telephone network which is shown to include a plurality of telephone exchanges, such as local exchanges 4 and transit exchanges 5, and trunks 6 there between. Telephone terminals 8 and 8' are connected to respective PSTN systems by means of exchanges 4 and 4' via telephone lines 9.

In a digital circuit switched telephone network the network signalling is typically, but not always, implemented by using common channel signalling system No. 7 (SS7 or C7) by ITU-T (International Telecommunication Union, telecommunication Standardization Sector), which is a global standard for telecommunications. SS7 defined the procedures and protocol by which network element in the PSTN exchange information over a digital signalling network to effect wireline and also wireless set-up, routing and control of calls. There are several national variants of the SS7, such as ANSI and Bellcore in the USA and ETSI in Europe.

The hardware and software functions of SS7 are divided in functional abstractions, i.e. levels or layers, which loosely correspond to OSI (Open System Interconnect) 7-layer model by ISO (the International Standards Organisation). Each of the layers is dedicated for a predefined purpose, see figure 2 disclosing the main functions of each layer. The first three layers contain a message transfer part (MTP). From these layer 1 defines the characteristics of the signalling link, layer 2 ensures accurate end-to-end transmission of a message across a signalling link and layer 3 provides message routing between signalling points in the SS7 network. As disclosed by figure 2, layers 4 to 7 contain ISDN User Part (ISUP) and/or Telephony User Part (TUP) and/or some application parts AP/TCAP (Application Part/Transaction Capabilities Application Part) and SCCP (Signalling Connection Control

Part) depending from the used dangling type.

An SS7 message is called a signal unit (SU) used to carry the signalling messages between different nodes of the telecommunication network. A SU contains three kinds of signal units: Fill-In Signal Units (FISU), Link Status Signal Units (LSSU) and Message Signal Units (MSU). From these MSUs carry all call control, database query and response, network management, and network maintenance data in a signalling information field (SIF). A MSU contains a service indicator and a routing label information which define implicitly the destination of a certain individual data packet. The MSU transfers also the signalling messages on the signalling links, said MSUs corresponding to standardised data packet formats for transfer and handling in the MTP (Message Transfer Part).

The MTP level 3 routes messages based on the routing label in the SIF of the MSU. The routing label comprises destination point code (DPC), originating point code (OPC), and signalling link selection field (SLS). The point codes are numeric which uniquely identifying each signaling point in the SS7 network. When the destination point code in a message indicates the receiving signaling point, the message is distributed to the appropriate user part indicated by the service indicator (SI) in the SIO. Messages destined for other signalling points are transferred provided that the receiving signalling point has message transfer capabilities. The selection of outgoing link is based on information in the DPC and SLS.

Figure 3a discloses the structure and main information content of one MSU. The SIF (signalling Information Field) and SIO (Service Information Octet) in the MSU contain information which are mandatory parameters of TUP (Telephony User Part). More precisely, the SIF includes the following information parameters: originating point code (OPC), destination point code (DPC), circuit identification code

(CIC). It is noted herein that CIC is necessary for ISUP/TUP only, but not required in SCCP. The SIO, in turn, includes parameters such as network indicator (NI) and service indicator (SI). As already explained, the OPC indicates the identity of the signalling point (SP) which is the sender of the signalling message and the DPC indicates the identity of the receiving SP. The CIC in turn indicates the number identity of a trunk connection (for speech, fax, data) to which a certain message belongs, and the time slot which is controlled by this MSU (i.e. the trunk circuit which is reserved by the originating exchange to carry the call). The NI of the SIO indicates the signalling network (national or international) to which a signalling message belongs.

An ISUP IAM (ISDN User Part; Initial Address Message) includes similar parameters, as is disclosed by figure 3b (see also figure 6). In the PSTN the IAM is sent in the forward direction by each switching point needed to complete a circuit between the two parties until the destination switching point is connected. An IAM contains the called party number in the mandatory variable part thereof. It may also contain the originating party name and number in the optional part thereof. (For more detailed information concerning ISUP messages, see e.g. ITU-T Recommendation Q.763).

Returning now back to figure 1, a data processing device 10 is disclosed to be connected to the Internet 2 through connection 14 in a manner known to the skilled person. The skilled person understands that there are several alternative ways for providing the Internet access, e.g. through a modem or an ISDN (Integrated Services Digital Network) connection or an Internet access server (IAS) connection, but since the method of accessing the Internet or similar packet switched data network is not an essential part of the invention, it will not be discussed herein in more detail.

The system is provided also with a gateway node or access

server 12 which interfaces the PSTN and the Internet. In this connection it is noted that even the networks can be disclosed as physically separated networks, the network arrangement can be such that the PSTN network cables are also used for the actual Internet data transmission, i.e. the PSTN acts as a bearer network for TCP/IP (Transport Control Protocol/Internet Protocol) traffic.

In the Internet the addressing is based on IP addresses in accordance with a TCP/IP (Transport Control Protocol/Internet Protocol) suite. The TCP/IP operates according to the principle of layered communication (cf. open systems integration i.e. the OSI model). TCP/IP communicates in a connectionless manner (meaning that no connections are set up through intermediate nodes, even if a logical connection is established between the client and the server). The layer-3 functions of the IP protocol make it possible to communicate over a number of interconnected networks. The network layer includes functions for routing and addressing. Without going to details, it is mentioned herein that the source and the destination are identified through using fixed length IP addresses included in the IP headers as explained in the background part of this specification.

Each Internet host computer and router has an IP address of its own. At present all IPv4 addresses have a length of 32 bits. The addresses are placed in the IP packet's source address and destination address fields. Only a portion of the address is used for the actual routing process, namely the network address. The IP addresses are divided into five classes A to E which are reserved to different purposes, the length of the network address being a function of the class. In decimals the form of an address for a computer is e.g. 130.100.75.19.

The communication system of figure 1 includes further a network signalling server (SS) 16. This network node contains all routing information for a call, such as the Originating

Point Code (OPC), Destination Point Code (DPC), Network Indicator (NI), Circuit Identification Code (CIC) and in some instances also a Requesting Country Code (RCC). This set of information provides a unique combination which can be compared with the IP address of a terminal communicating through the IP network. In other words it is possible to map the IP address on basis of the relation between the combination of the routing signalling parameters and the IP addresses. This address relation, which can be implemented in form of a mapping table, can be allocated in some of the existing nodes of the network (for instance, in the Intelligent Network (IN) of the PSTN or PLMN) or then it can be implemented by the provision of the new server device 16. In general, the arrangement can be such that when a call is established from the terminal 8, required signalling information parameters for the address conversion are provided to the signalling server 16 which then in response provides the IP address.

The server 16 can be local or global. In case any updates are made to the address information or to the above mentioned parameters, the server 16 will be informed accordingly.

Figure 4 discloses more precisely an example of the table implemented in the signalling server 16. The IP address and corresponding combinations of the routing information parameters are mapped into the table such that the appropriate pairs are found and retrieved on request, whereby the call can be routed to the desired address having a different format than what the originating terminal uses.

Now, when a user of a telephone terminal 8 connected to the PSTN 1 wants to establish a call to a terminal beyond the Internet, such as to an Internet terminal 10 connected to an exchange of the bearer network by means of an ISDN connection or to the ordinary telephone terminal 8', the user can initiate the procedure by dialling in a telephone number (B-number) of the desired destination terminal. The number may

comprise a special prefix or it can be otherwise formulated such that the network apparatus will recognise that this call is not an ordinary PSTN call, but that the call is to be routed to the desired terminal via the packet switched Internet. The destination terminal can have an IP address instead of a conventional telephone number or it may be an ordinary telephone having a normal B-number.

The signalling is directed to a gateway or signalling server 16 including the routing table and controlling the signalling operations in accordance with the principles of the present invention. Even though it is necessary to have some kind of access node, e.g. the access server 12 of figure 1, between the networks so that the user may access the IP network, it is not necessary for the signalling to go directly to the access node from the exchange, but it is possible to have the gateway or signalling server between the access node and the exchange. The actual data transmission and the speech path go directly to the destination exchange but the signalling may be routed through a different route, as is also disclosed by figure 1.

The signalling server associates the predefined combination of the OPC, DPC, NI, CIC and RCC parameters with the desired IP address in the table. This IP address enables then the access to the TCP/IP network, and the connection will be routed by means of routers of the data network in accordance with the IP addressing protocols to the desired IP address.

In case the receiving party is an ordinary telephone terminal, the IP-connection is routed and then terminated to a local exchange of the receiving B-subscriber (see the flow chart of figure 5). The B-number information may have been included in the IP header and is thus received together with the data packet, whereafter the local exchange of the B-subscriber routes the call to the B-subscriber using the B-number information of the telephone network.

Thus, the invention provides an apparatus and a method by which a significant improvement can be achieved in the area of routing calls between different networks. It should, however, be noted that the foregoing examples of the embodiments of the invention are not intended to restrict the scope of the invention to the specific forms presented above but the present invention is meant rather to cover all modifications, similarities and alternatives which are included in the spirit and scope of the present invention, as defined by the appended claims.

Claims

1. A method of setting up a connection between a first point within a circuit switched network and a second point within a packet switched network, wherein data is routed in the circuit switched network using signalling information and data is routed in the packet switched network using data network addresses, comprising steps

providing a translation table containing signalling information for a set of unique connections within the circuit switched network and respective data network addresses in the packet switched network;

for a connection initiated from said first point to said second point using signalling information, converting the signalling information to a data network address using said translation table, and

for a connection initiated from the second point to the first point using a data network address, converting the data network address into signalling information using the translation table.

2. Method in accordance with claim 1, wherein signalling information parameters include at least one of the parameters of signalling identification field (SIF) in a message signalling unit (MSU), preferably at least one of originating point code (OPC), destination point code (DPC), circuit identification code (CIC), network indicator (NI), requesting country code (RCC).

3. Method in accordance with claim 2, wherein the signalling information consist of a combination of originating point code (OPC), destination point code (DPC), circuit identification code (CIC) and network indicator (NI).

4. Method in accordance with any of the preceding claims, wherein the packet switched data network is the Internet, the addresses thereof being formed in accordance with IP address protocol.

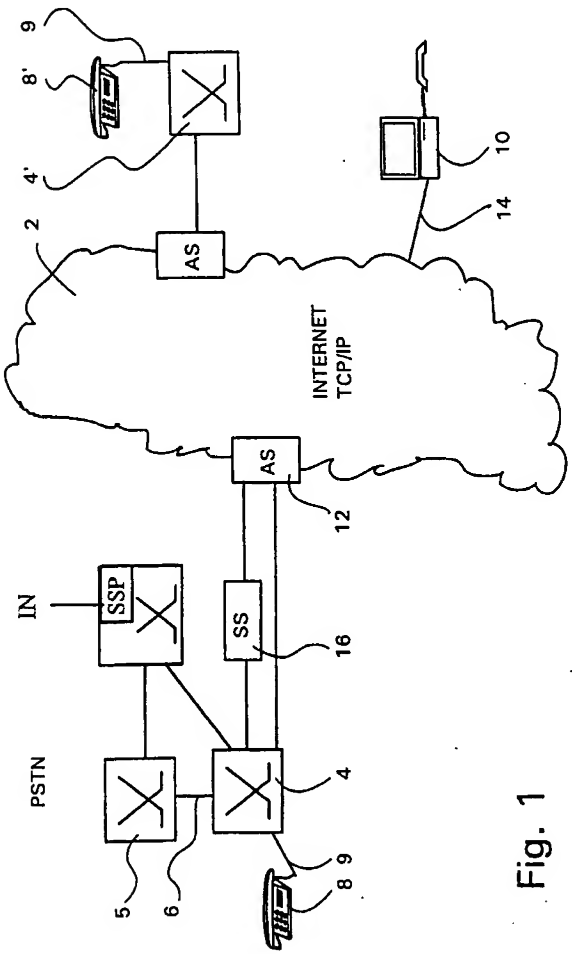

Fig. 1

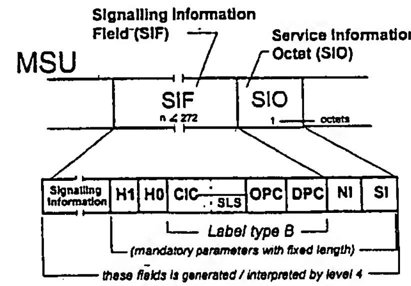

Fig. 3a

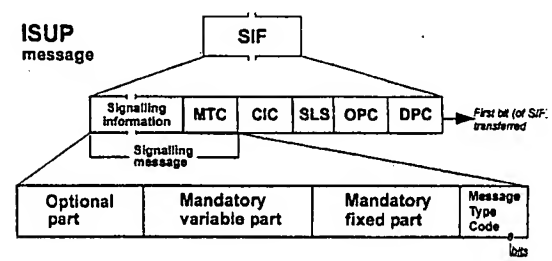

Fig. 3b

5. Method in accordance with any of the preceding claims, comprising storing of the signalling parameters and the data network address information in a signalling server and accomplishing the comparing of routing parameters and the address information in the signalling server.

6. Method in accordance with claim 5, wherein the call establishment signalling is routed to the signalling server and the speech path is routed directly to the destination exchange.

7. Method in accordance with any of the preceding claims, wherein the connection is routed between a first telephone terminal connected to a public switched telephone network and a second telephone terminal connected to another public switched telephone network via a packet switched data network.

8. A telecommunication system comprising:

a circuit switched network wherein the routing of data is based on routing information parameters;

a packet switched network wherein the routing of data is based on data network addresses;

an access node between the networks;

a signalling node including a record of the routing information parameters forming unique combinations for a set of connections and respective data network addresses, wherein the signalling node is arranged to associate the unique combinations of the routing information parameters with the data network addresses for obtaining a desired destination point for a call made from the circuit switched network to the packet switched network or vice versa.

9. A system in accordance with claim 8, wherein the unique combination of the signalling parameters includes at least one of the parameters of signalling identification field (SIF) in a message signalling unit (MSU), preferably at least one of originating point code (OPC), destination point code (DPC), circuit identification code (CIC), network indicator (NI), and requesting country code (RCC).

10. A system in accordance with claim 8 or 9, wherein the signalling node is a signalling server or signalling gateway implemented in connection with a local telephone exchange.

11. A system in accordance with any of claims 8 to 10, wherein the packet switched network is the Internet.

12. A signalling server, including

a translation table containing signalling information for a set of unique connections within a circuit switched telecommunication network, and respective data network addresses in a packet switched network, wherein the signalling server is arranged to, for a connection initiated from a first point to a second point using signalling information, convert the signalling information to a data network address using said translation table, and, for a connection initiated from the second point to the first point using a data network address, convert the data network address into signalling information.

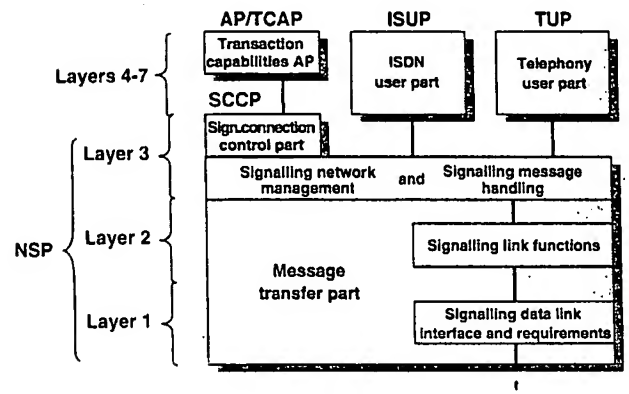

Fig. 2

Routing information /SIF					IP address
OPC	DPC	NI	CIC	RCC	

Fig. 4

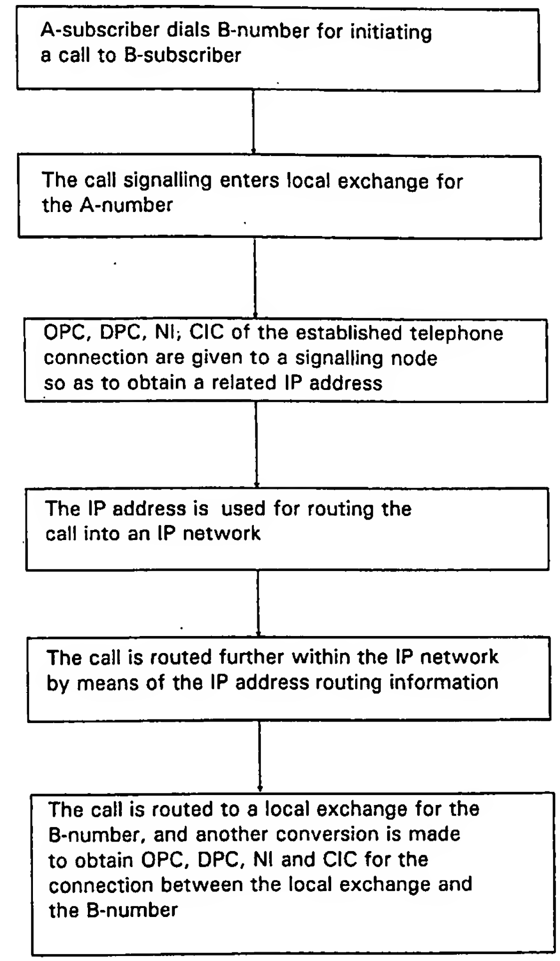

Fig.5

ISUP signalling sequence:

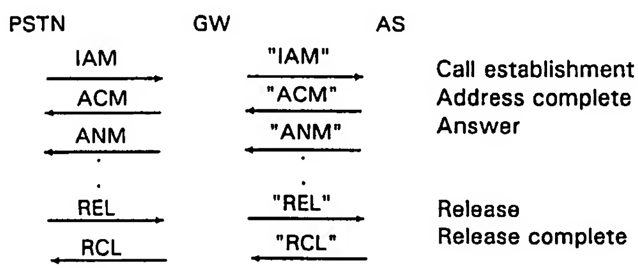

Fig. 6